Multimedia Service Platform

Turnkey solution for SIP based communications

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Current telecommunications landscape

- From the old PSTN only the E.164 numbering plan remains
- SS7 and ISDN (circuit switched) networks reached their end-of-life
- In the end it will be one “all IP” global network, the Internet
- Replacement for signaling protocols has been proposed by IETF and ITU, first one was H323 (ITU) and SIP came second (IETF)
- ENUM complements SIP and allows IP devices to be reached from the PSTN without programming switches with numbering plans
- Alternatives are H323, MGCP/Megaco,Skype, Skinny or IAX.

All vendors support SIP today, it is a common denominator in the vendor landscape

SIP - the Session Initiation Protocol

- SIP is a horizontal, it allows end-points to find each other and initiate sessions over the Internet (any type of sessions)
- Intelligence is distributed among participating nodes, new applications can be rolled out without upgrades in the network
- SIP makes use of existing proven protocols for addressing, security and scalability, thus it did not re-invent the wheel (DNS, HTTP, TLS)
- NAT traversal has been addressed in SIP (not H323)
- SIP coexists happy with all other protocols (all vendor support)
- SIP replaces telephone numbers with e-mail addresses

SIP enabled Convergence between fixed and mobile networks, you can change access media, device or provider and it still works

Convergence and NGN

- Convergence means from Telecom + Internet we end up with one network alone, NGN is the Internet
- Mobile fixed or cable converge into the services at the end-user
- Internet is the dumb network the service is at the edge and not in the center
- The end-user is the center and not the carrier/provider
- You stay in business as long as your service is interesting enough
- You cannot build walled gardens anymore

This means new business models need to be developed

Open garden versus Walled garden

- Walled gardens so far have been motivated by poor implementations, lack of clarity over new business models, the push of Session Border Controllers
- PSTN and E164 numbering plan were a success because one was able to connect to everybody else (universal service)
- Is important to allow dialing of both E164 numbers and SIP URIs

“By the time you finish building up your walled garden, the customers will be safely outside” - Adrian Georgescu

“If you can dial a SIP URI and you can be called by a SIP URI than you have access to VoIP otherwise is just an emulation of classic PSTN” - Henry Sinnreich

Exiting business models based on SIP

SIP enables several selling points today:

- PSTN termination service with per minute charging
- ADSL service complement
- SIP devices
- Telephone numbers sold separately
- Company IP PBXs interconnected via IP
- SIP service subscription

But PSTN services will be short lived services, you must change a bit the mentality

Exiting business models based on SIP

The current approach of VoIP service providers is wrong:

- They compete to reach to bottom, till price per minute equals zero
- Incorrectly marketing VoIP as free, while somebody has to pay for it
- Applying telco logic to Internet does not work (see IMS design)
- Build walled gardens while the end-user may opt-out from its device
- Few telco comes out with innovative services
- They do not provide end-user with a SIP address, but only a classic telephone number

When everybody left the PSTN and VoIP is free where is the cake?

New business models based on SIP

SIP is more than Voice over IP. New services will emerge some of them you will have never heard before of:

- Mobility
- Presence and contextual communications
- Notifications based on business triggers
- Identify and certificate management, privacy services
- Synchronize data among devices
- Address translation (ENUM and domain names)

The cake is there you just need an enabler and that is SIP

Ingredients

Operators deploying SIP must address the following:

- SIP signaling - SIP Proxy / Registrar / Redirect
- E164 numbering resources and interconnect with the PSTN
- Far-end NAT traversal capability for devices behind NAT
- CDR mediation, accounting and tracing
- Provisioning interfaces
- Number portability and peering with other SIP providers
- Emulation of existing telephony services (class 5)
- Applications (Voicemail, Conferencing, IM and Presence)
- End-user devices (soft phones, hard phones)

Implementation scenarios

Operators may chose between:

1. Collocated: Implement infrastructure from scratch, either trough an integrator or direct vendor selection (2 to 6 months)

2. Hosted: Find a white-label technology vendor and use a hosted service), this approach is good to gain time to market while implementing 1

3. Developed: Hire skilled engineers and develop systems based on Open Source components (6-12 months), this approach has unpredictable results

Requirements

When selecting the vendors there are key issues that must be addressed by the service provider:

- Compliancy with standards
- Support industry best practices (Trials, interoperability)
- Knowledge base
- Solution scalability
- Integration with other systems
- Availability of source code and debugging tools

The proposition

One solution for implementing SIP services using collocated, hosted or a combined model

- Complete solution directly from the vendor (predictable end-result)
- Based on proven components developed in Open Source model (large knowledge base)
- Trials during 2003, enjoys production status since 2004
- Low maintenance cost (one FTE can run and maintain the platform)

Commercial SIP services are now available based on it:

@Call (ARCOR), Talkin2YA (Budget Phone Company), Eurovoice (Euroweb International), SIP2GO (Sentiro)

Components

Multiple components work together smoothly to achieve the goal

- SIP signaling: OpenSER
- NAT traversal: MediaProxy
- E.164 number translation and number portability: ENUM
- Mediation and accounting: CDRTool
- Accounting: FreeRadius
- Database: MySQL
- Provisioning system: NGN-Pro (SOAP/XML API)
- Voicemail and voice to email: Asterisk

The goal is subscriber in, invoice out

Deployment track

During installation the Operator can learn how to build the platform, which is useful for training or disaster recovery purposes

The following items are delivered with a standard installation:

- Installation log
- Source code
- Documentation for individual components
- As-built schematic (platform blueprint)
- Sample client for remote SOAP/XML provisioning

Managed services

AG Projects hosted facility provide fast deployment of SIP services

- SIP audio, video and IM
- White-label VoIP platform - all included (SIP signaling, NAT traversal, ENUM, provisioning and accounting), time to go to market is minimum (rollout in 2-4 weeks)
- Work with external PSTN gateways (like Cisco) or SIP termination carriers (like MCI or Level 3) or your SIP PBX (Asterisk)
- All you need is to print your own invoices

Provisioning interfaces

The following functions are available out-of-the box on the hosted platform:

- Add/modify/delete customers
- Add/modify/delete SIP accounts
- Add/modify/delete DNS zones
- Add/modify/delete ENUM mappings
- Customizable SIP end-user control panel (SIP settings page) where end-users can login and access their own account
- Access to Call Detail Records, SIP traces and rating engine

SOAP/XML provisioning engine allows the development of web sites hosted outside the platform

Scalability

Multimedia Service Platform is able to grow with your business needs:

- Multimedia Service Platform is designed for operators with maximum 50,000 subscribers
- SIP Thor is the upgrade path to scale to millions of subscriber

SIP Thor is AG Projects solution for survivability and scalability of SIP communications to serve a base of multi-million subscribers.

SIP Thor provides a more flexible and lower cost alternative to the IMS concept proposed by 3GPP and enables next generation SIP services like user mobility, video, IM and Presence.

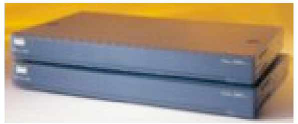
Supported SIP devices

All SIP compliant software and hardware phones are supported

ATA adaptors

Cisco ATA

Asterisk

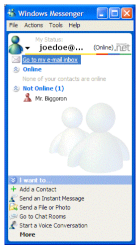


PSTN gateways

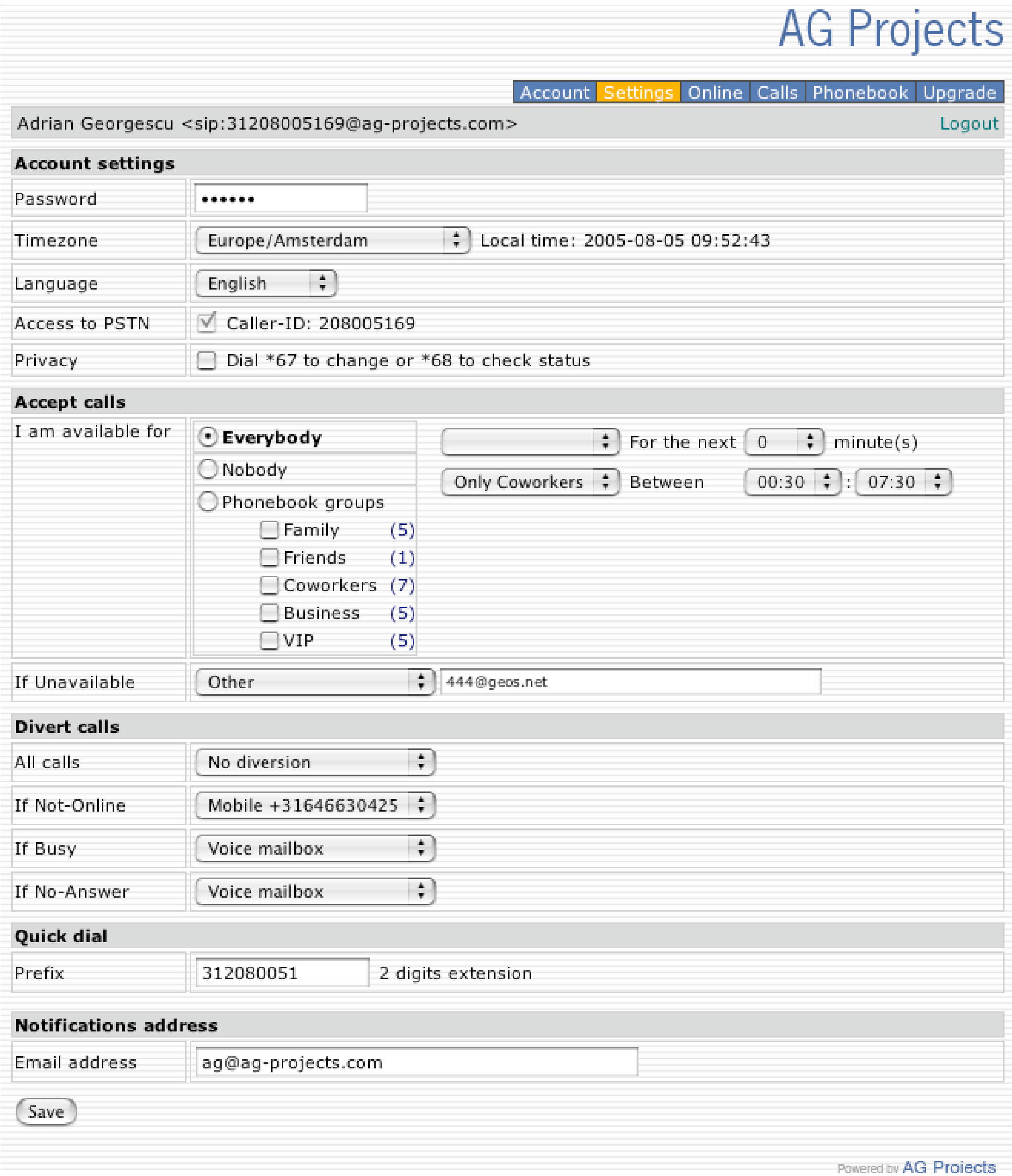
Video Phones

Desktop phones

Software phones



WiFi Phones



SIP settings page

Call forwarding
Do not disturb
Go to meeting
Time based forwarding
Short dial codes
Privacy control
Selective call accept
Selective call reject
Show online devices
Show last calls
Phonebook
Voicemail settings

Access to Call Detail Records, traces and rating

CDRTool
Powered by AG Projects

SIP Express Router (Global Switch)

Call search | Rating tables| Log| Accounts | 2005-08-05 10:08:50 (Europe/Amsterdam) | CDRTool 3.2.9

Logged in as **adriang (Adrian Georgescu)** Logout

Refine search | Refresh | Export results to file | Save a description for this query: Save

105 records found.

Found 6 CDRs for normalization.

From 2005-08-05 09:08 to 2005-08-05 23:55

	Session start time	SIP caller	In	SIP destination	Out	Dur	Price	KBIn	KBOut	Status	Codecs
1N	2005-08-05 10:08:02	0613159157@voipgw01.budgetphone.nl	In	31747110327@budgetphone.nl	Out	00:00		0.00	0.00	InternalServerError (500)	
2N	2005-08-05 10:07:22	2460306@sip2go.com	In	+919892546052 (India mobiel 9198)		00:00		0.00	0.00	Canceled (487)	
3N	2005-08-05 10:06:19	31162517145@budgetphone.nl	In	+31115567366 (Nederland 31)	Out	00:00		0.00	0.00	Canceled (487)	
4N	2005-08-05 10:06:08	31717110340@budgetphone.nl	In	+31653779120 (Nederland mobiel 31653)	Out	00:52	0.2010	457.80	448.55	Ok (200)	GSM

Signalling information
SIP Session: 1AC66B29-1A10-44B7-945F-EE8E556E57DB@83.85.113.250
SIP Method: Invite from *83.85.113.250:5060*
SIP Status: Ok (200) DelayTime: 11(s)
SIP From: sip:31717110340@budgetphone.nl
SIP To: sip:0031653779120@budgetphone.nl
PSTN Caller ID: 717110340 Privacy disabled
Start time (caller): 2005-08-05 10:06:08 Europe/Amsterdam
Start time (proxy): 2005-08-05 10:06:08 Europe/Amsterdam
Stop time (proxy): 2005-08-05 10:07:00 Europe/Amsterdam
Session duration: 00:52
SIP Proxy: 81.23.228.139
SIP Canonical URI: sip:0031653779120@voipgw01.budgetphone.nl
Next SIP hop: sip:0031653779120@voipgw01.budgetphone.nl
Destination name: Nederland mobiel
Billing Party: 31717110340@budgetphone.nl

Media information
Application: Audio
Codecs: GSM
Caller UA: Talkin 2 Ya release 1103m
Called UA: Asterisk PBX

Rating information
ConnectFee: 0.0450
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Span: 1
Duration: 52 s
Appl: PSTN voice (906 KB)
Destination: 31653 (Nederland mobiel)
Customer: domain=budgetphone.nl
StartTime: 2005-08-05 10:06:08 (Europe/Amsterdam)
ProfileId: 441 for weekday
RateId: 441 for 8-19h
Rate: 0.1800 / 60 s
Price: 0.1560

	Session start time	SIP caller	In	SIP destination	Out	Dur	Price	KBIn	KBOut	Status	Codecs
5N	2005-08-05 10:04:44	31237110322@budgetphone.nl	In	+31534302876 (Nederland 31)	Out	00:43	0.0308	1,019.92	1,090.59	Ok (200)	G711u
6N	2005-08-05 10:04:13	0434571887@voipgw01.budgetphone.nl	In	31437110334@budgetphone.nl		00:00		0.00	0.00	NotFound (404)	
7N	2005-08-05 10:03:53	0434571887@voipgw01.budgetphone.nl	In	31437110334@budgetphone.nl		00:00		0.00	0.00	NotFound (404)	
8N	2005-08-05 10:03:13	31107110332@budgetphone.nl	In	+31627284919 (Nederland mobiel 31627)	Out	00:43	0.1740	247.45	252.54	Ok (200)	G729
9N	2005-08-05 10:02:18	2460306@sip2go.com	In	+919892546052 (India mobiel 9198)		00:00		0.00	0.00	ServiceUnavailable (503)	
10N	2005-08-05 10:02:09	31307110365@budgetphone.nl	In	+31206485016 (Nederland 31)	Out	00:29	0.0273	649.14	700.78	Ok (200)	G711u
11N	2005-08-05 10:01:07	2460306@sip2go.com	In	+912228845554 (India (Bombay) 9122)		00:44	0.1833	268.36	352.38	Ok (200)	G729

Management of subscribers

One central place to manage any type of record (DNS, SIP, ENUM, Email)

| Accounts | Order | Records | Zones | Domains | Support | Help

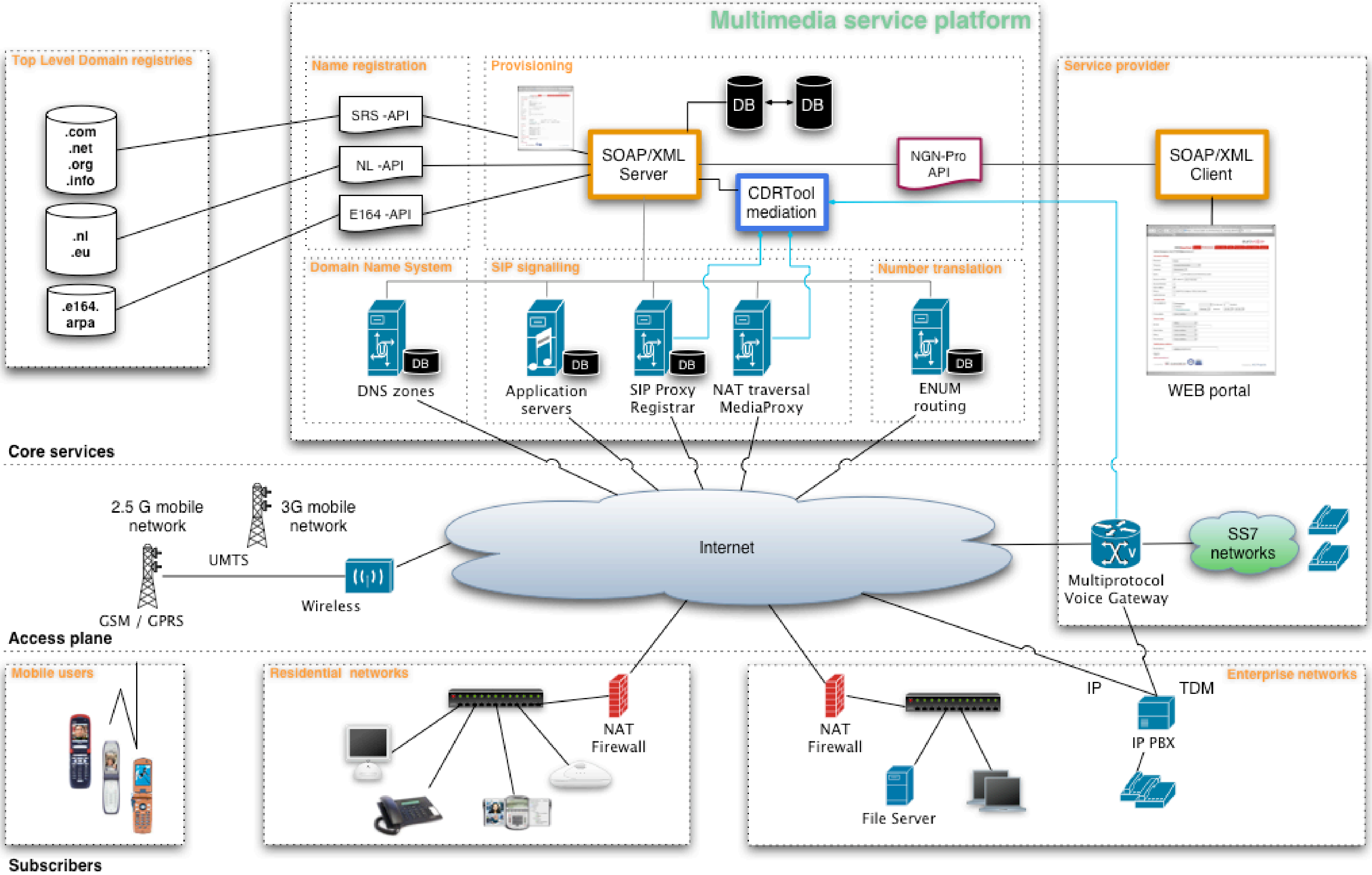
Logged in as **Adrian Georgescu** Logout

Type: SIP accounts · All accounts · All record types · Order by: Change date · Desc · Search

1	Update	NAPTR	Number	+878102233344209	Forward to	SIP	sip:2233344209@sip2sip.info	Pref	0	Owner 2983	Delete
2	Update	SIP	Name	Mark Hewitt	Account		2233344209@sip2sip.info	Password	•••••	Owner 2983	Delete
3	Update	SIPA	Address	mark.hewitt@sip2sip.info	Forward to		2233344209@sip2sip.info			Owner 2983	Delete
4	Update	NAPTR	Number	+878102233344208	Forward to	SIP	sip:2233344208@sip2sip.info	Pref	0	Owner 2982	Delete
5	Update	SIP	Name	demuregoat goaty	Account		2233344208@sip2sip.info	Password	•••••	Owner 2982	Delete
6	Update	SIPA	Address	demuregoat.goaty@sip2sip.info	Forward to		2233344208@sip2sip.info			Owner 2982	Delete
7	Update	NAPTR	Number	+878102233344207	Forward to	SIP	sip:2233344207@sip2sip.info	Pref	0	Owner 2970	Delete
8	Update	SIP	Name	xxxx xxxxxx	Account		2233344207@sip2sip.info	Password	•••••	Owner 2970	Delete
9	Update	SIPA	Address	alex.buss@sipline.nl	Forward to		2233344040@sipline.nl			Owner 2969	Delete
10	Update	NAPTR	Number	+878102233344040	Forward to	SIP	sip:2233344040@sipline.nl	Pref	0	Owner 2969	Delete
11	Update	SIP	Name	alex buss	Account		2233344040@sipline.nl	Password	•••••	Owner 2969	Delete
12	Update	SIP	Name	Michiel Leenaars	Account		2233344197@sipline.nl	Password	•••••	Owner 2939	Delete
13	Update	NAPTR	Number	+31208005134	Forward to	SIP	sip:2233344197@sipline.nl	Pref	0	Owner 2939	Delete
14	Update	EMAILF	Address	program@agnext.com	Forward to		toast@example.com			Owner 0	Delete

Platform blueprint

Multimedia service platform



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This presentation is available at:

http://ag-projects.com

Questions?

Thank you,

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